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Please find below and/or attached an Office communication concerning this application or proceeding.

Office Action Summary	Application No. 09/967,062	Applicant(s) ALLRED ET AL.	
	Examiner Lun-See Lao	Art Unit 2643	

-- *The MAILING DATE of this communication appears on the cover sheet with the correspondence address* --

Period for Reply

A SHORTENED STATUTORY PERIOD FOR REPLY IS SET TO EXPIRE 3 MONTH(S) FROM THE MAILING DATE OF THIS COMMUNICATION.

- Extensions of time may be available under the provisions of 37 CFR 1.136(a). In no event, however, may a reply be timely filed after SIX (6) MONTHS from the mailing date of this communication.
- If the period for reply specified above is less than thirty (30) days, a reply within the statutory minimum of thirty (30) days will be considered timely.
- If NO period for reply is specified above, the maximum statutory period will apply and will expire SIX (6) MONTHS from the mailing date of this communication.
- Failure to reply within the set or extended period for reply will, by statute, cause the application to become ABANDONED (35 U.S.C. § 133). Any reply received by the Office later than three months after the mailing date of this communication, even if timely filed, may reduce any earned patent term adjustment. See 37 CFR 1.704(b).

Status

1)☒ Responsive to communication(s) filed on *25 February 2004*.

2a)☐ This action is **FINAL**. 2b)☒ This action is non-final.

3)☐ Since this application is in condition for allowance except for formal matters, prosecution as to the merits is closed in accordance with the practice under *Ex parte Quayle*, 1935 C.D. 11, 453 O.G. 213.

Disposition of Claims

4)☒ Claim(s) *1-20* is/are pending in the application.

4a) Of the above claim(s) _____ is/are withdrawn from consideration.

5)☐ Claim(s) _____ is/are allowed.

6)☒ Claim(s) *1-20* is/are rejected.

7)☐ Claim(s) _____ is/are objected to.

8)☐ Claim(s) _____ are subject to restriction and/or election requirement.

Application Papers

9)☐ The specification is objected to by the Examiner.

10)☐ The drawing(s) filed on _____ is/are: a)☐ accepted or b)☐ objected to by the Examiner.

Applicant may not request that any objection to the drawing(s) be held in abeyance. See 37 CFR 1.85(a).

Replacement drawing sheet(s) including the correction is required if the drawing(s) is objected to. See 37 CFR 1.121(d).

11)☐ The oath or declaration is objected to by the Examiner. Note the attached Office Action or form PTO-152.

Priority under 35 U.S.C. § 119

12)☐ Acknowledgment is made of a claim for foreign priority under 35 U.S.C. § 119(a)-(d) or (f).

a)☐ All b)☐ Some * c)☐ None of:

1.☐ Certified copies of the priority documents have been received.

2.☐ Certified copies of the priority documents have been received in Application No. _____.

3.☐ Copies of the certified copies of the priority documents have been received in this National Stage application from the International Bureau (PCT Rule 17.2(a)).

* See the attached detailed Office action for a list of the certified copies not received.

Attachment(s)

1) ☒ Notice of References Cited (PTO-892)

2) ☐ Notice of Draftsperson's Patent Drawing Review (PTO-948)

3) ☐ Information Disclosure Statement(s) (PTO-1449 or PTO/SB/08) Paper No(s)/Mail Date _____.

4) ☐ Interview Summary (PTO-413) Paper No(s)/Mail Date. _____ .

5) ☐ Notice of Informal Patent Application (PTO-152)

6) ☐ Other: _____.

DETAILED ACTION

Introduction

1. This action is response to amendment filed on 02-25-2004. Claims 1, 4,7,11, 15 and 16 have been amended. Claims 1-20 are pending.

Double Patenting

2. The nonstatutory double patenting rejection is based on a judicially created doctrine grounded in public policy (a policy reflected in the statute) so as to prevent the unjustified or improper timewise extension of the "right to exclude" granted by a patent and to prevent possible harassment by multiple assignees. See *In re Goodman*, 11 F.3d 1046, 29 USPQ2d 2010 (Fed. Cir. 1993); *In re Longi*, 759 F.2d 887, 225 USPQ 645 (Fed. Cir. 1985); *In re Van Ornum*, 686 F.2d 937, 214 USPQ 761 (CCPA 1982); *In re Vogel*, 422 F.2d 438, 164 USPQ 619 (CCPA 1970);and, *In re Thorington,* 418 F.2d 528, 163 USPQ 644 (CCPA 1969).

A timely filed terminal disclaimer in compliance with 37 CFR 1.321(c) may be used to overcome an actual or provisional rejection based on a nonstatutory double patenting ground provided the conflicting application or patent is shown to be commonly owned with this application. See 37 CFR 1.130(b).

Effective January 1, 1994, a registered attorney or agent of record may sign a terminal disclaimer. A terminal disclaimer signed by the assignee must fully comply with 37 CFR 3.73(b).

3. Claims 1-20 are rejected under the judicially created doctrine of obviousness-type double patenting as being unpatentable over claims 1-16 of U.S. Patent No. 6,721,428. Although the conflicting claims are not identical, they are not patentably distinct from each other.

Consider claim 1 substantially all the claimed steps were claimed in the patent identified above, such as the steps of : "providing second digital data representing an initial response curve of an initial hearing ability to be enhanced of sound level versus frequency;

comparing aid first digital data to said second digital data and determining whether said initial response curve is within said tolerance range;

if said initial response curve is not within said tolerance range, iteratively generating digital filter coefficients controlling center frequency, filter bandwidth and amplitude for a succession of additional digital audio filters, applying all currently generated digital audio filters to said second digital data to generate third digital data for a compensated response curve, and automatically optimizing the center frequency, amplitude and filter bandwidth of said currently generated digital audio filters until said compensated response curve is within said tolerance range or a predetermined limit on the number of digital audio filters has been reached, whichever occurs first" (see US PAT. 6,721,428 claim 1, lines 3-21).

The difference between the current claims and the patent is that the environments wherein the claimed digital filter in used the use of the claimed digital filter. The current application involve a hearing aid and the patent involves loudspeaker.

However, Both hearing aid and loudspeaker are involving an operation or turning of a speaker. This digital filter is meant to use with a speaker. Although the environment (one is hearing aid, the other one is for loudspeaker) is different, but they both involve speakers. Therefore, using this digital filter in these two environments would have been obvious for one of ordinary skill in the art because even the environment changes, the operation of the digital filter remains substantially unchanged.

Consider claim 4 substantially all the claimed steps were claimed in the patent identified above, such as the steps of : " providing first digital data for a tolerance range for a target response curve representative of said enhanced hearing ability of sound level versus frequency;

providing second digital data representative of an initial response curve of an initial hearing ability to be enhanced of sound level versus frequency;

comparing said first digital data to said second digital data and determining whether said initial response curve is within said tolerance range; and

if said initial response curve is not within said tolerance range, generating a set of digital filter coefficients controlling center frequency, filter bandwidth and amplitude for a succession of digital audio filters to tune said hearing aid by performing the following optimizing steps iteratively,

digitally processing said second digital data to determine an n^{th} set of initial digital filter coefficients for an nth digital filter for an n^{th} peak in said actual initial curve where said initial response curve is not within said tolerance range, including a center

frequency, and an amplitude and a bandwidth for said peak, where n^{th} is the number of an iteration of said optimizing steps,

digitally generating digital filter coefficients controlling center frequency, filter bandwidth and amplitude for a compensating n^{th} digital filter from said n^{th} set of initial parameters,

applying said n^{th} digital filter to said second digital data and modifying said n^{th} set of initial parameters to determine an n^{th} set of optimum digital filter coefficients for said compensating n^{th} digital filter, to generate third digital data for an n^{th} interim compensated response curve of sound level versus frequency,

processing said third digital data to determine whether said n^{th} interim compensated response curve is within said tolerance range,

if said n^{th} interim compensated response curve is not within said tolerance range, performing another iteration of said optimizing steps until said interim compensated response curve is within said tolerance range or a predetermined limit on the number of digital filters has been reached, whichever occurs first (see US PAT. 6,721,428 claim 5, col. 11 line 30-col.12 line 2).

The difference between the current claims and the patent is that the environments wherein the claimed digital filter in used the use of the claimed digital filter. The current application involve a hearing aid and the patent involves loudspeaker.

However, Both hearing aid and loudspeaker are involving an operation or turning of a speaker. This digital filter is meant to use with a speaker. Although the environment (one is hearing aid, the other one is for loudspeaker) is different, but they both involve

speakers. Therefore, using this digital filter in these two environments would have been obvious for one of ordinary skill in the art because even the environment changes, the operation of the digital filter remains substantially unchanged.

Consider claim 7 substantially all the claimed steps were claimed in the patent identified above, such as the steps of : " providing first digital data for a tolerance range for a target response curve representative of said enhanced hearing ability of sound level versus frequency;

providing second digital data for an initial response curve of said hearing ability to be enhanced of sound level versus frequency;

comparing said first digital data to said second digital data and determining whether said initial response curve is within said tolerance range; and

applying said nth digital filter to said second digital data and modifying said n^{th} set of initial filter coefficients to determine an n^{th} set of optimum parameters for said n^{th} digital filter, to generate third digital data for an n^{th} interim compensated response curve of sound level versus frequency;

if n > 1, performing the following joint filter optimizing steps iteratively and cyclically, generating fourth digital data for interim computed response curves in which for each joint filter optimizing iteration one of said n filters is absent, and then performing said single filter optimization steps utilizing said fourth digital data to generate fifth digital data for an updated interim response curve, digitally processing said fifth digital data to determine whether the most recent of said joint filter optimizing iterations has resulted in

a change in said updated interim response curve greater than a predetermined amount of change, and if so continuing to perform said joint filter optimizing steps; processing said fifth digital data to determine whether said n^{th} interim compensated response curve is within said tolerance range, and if not,

performing another iteration of the foregoing steps until said interim compensated response curve is within said tolerance range or a predetermined limit on the number of digital filters has been reached, whichever occurs first, but if so, ceasing performance of further iterations, and

if said initial response curve is not within said tolerance range, generating a set digital filter coefficients controlling center frequency, filter bandwidth and amplitude for a succession of additional compensating digital audio filters by performing the following single filter optimizing steps iteratively, and digitally processing said second digital data to determine an n^{th} set of initial parameters for an n^{th} peak in said initial response curve where said initial response curve is not within said tolerance range, including a center frequency, an amplitude and a bandwidth for said peak, where n is the number of an iteration of said optimizing steps, digitally generating a compensating n^{th} digital filter from said n^{th} set of initial parameters (see US PAT. 6,721,428 claim 7 and col. 12 lines 6-59).

The difference between the current claims and the patent is that the environments wherein the claimed digital filter in used the use of the claimed digital filter. The current application involve a hearing aid and the patent involves loudspeaker.

However, Both hearing aid and loudspeaker are involving an operation or turning of a speaker. This digital filter is meant to use with a speaker. Although the environment (one is hearing aid, the other one is for loudspeaker) is different, but they both involve speakers. Therefore, using this digital filter in these two environments would have been obvious for one of ordinary skill in the art because even the environment changes, the operation of the digital filter remains substantially unchanged.

Consider claim 10 substantially all the claimed steps were claimed in the patent identified above, such as the steps of : "fitting said hearing aid to said individual;

connecting said hearing aid to a source of audio digital signals;

providing said individual with a device to generate indication signals at will;

receiving said indication signal during said generation of a signal of a selected frequency indicative of said individual hearing said selected frequency;

providing a digital computer connected to receive said first series of audio digital signals to generate digital data representative of said individual's hearing ability using said hearing aid without filters determined from said first series of digital signals computer programmed to determine said digital filter coefficients for digital filters for tuning said hearing aid and providing said coefficients to said digital audio processing unit in said hearing aid; and

generating and providing a first series of audio digital signals to said hearing aid, each digital signal in said first series corresponding to an analog audio signal having a selected frequency and multiple power levels;

at said hearing aid converting each of said series of digital signals into said corresponding analog audio signal;

providing a digital audio processing unit in said hearing aid for processing received audio digital signals corresponding to analog audio signals and providing processed audio digital data, including applying digital audio filters or tuning said hearing aid characterized by generating digital filter coefficients in algorithms applied to said received audio digital signals to effect said digital audio filters (see US PAT. 6,721,428 claim 9 and col.12 line 63-col.13 line 38).

The difference between the current claims and the patent is that the environments wherein the claimed digital filter in used the use of the claimed digital filter. The current application involve a hearing aid and the patent involves loudspeaker.

However, Both hearing aid and loudspeaker are involving an operation or turning of a speaker. This digital filter is meant to use with a speaker.

Although the environment (one is hearing aid, the other one is for loudspeaker) is different, but they both involve speakers. Therefore, using this digital filter in these two environments would have been obvious for one of ordinary skill in the art because even the environment changes, the operation of the digital filter remains substantially unchanged.

Consider claim 14 substantially all the claimed steps were claimed in the patent identified above, such as the steps of : " a source of first audio digital data corresponding to analog audio signals having a selected frequency and multiple power levels;

a digital to analog converter receiving said digital data from said digital audio processing unit and converting said processed digital data into a corresponding analog audio signal; and a speaker receiving said analog audio signal from said digital to analog converter and producing corresponding sound to the individual;

a device for generating indication signals indicative of said individual receiving said sound; and

a digital computer connected to receive said first audio digital data and said indication signals, said digital computer programmed to determine said digital filter coefficients for digital filters for tuning said hearing aid and provide said coefficients to said digital audio processing unit: and

a digital audio processing unit in said hearing aid for processing said first audio digital data according to at least one digital filter having digital filter coefficients controlling filter center frequency, amplitude and filter bandwidth and providing processed audio digital data, including applying digital audio filters for tuning said hearing aid characterized by coefficients in algorithms applied to said first audio digital data to effect said digital audio filters (see claim 11 and col.13 line 44-col.14 line 18).

The difference between the current claims and the patent is that the environments wherein the claimed digital filter in used the use of the claimed digital filter. The current application involve a hearing aid and the patent involves loudspeaker.

However, Both hearing aid and loudspeaker are involving an operation or turning of a speaker. This digital filter is meant to use with a speaker.

Although the environment (one is hearing aid, the other one is for loudspeaker) is different, but they both involve speakers. Therefore, using this digital filter in these two environments would have been obvious for one of ordinary skill in the art because even the environment changes, the operation of the digital filter remains substantially unchanged.

Consider claim 16 substantially all the claimed steps were claimed in the patent identified above, such as the steps of : " providing first digital data for a tolerance range for a target response curve representative of said enhanced hearing ability of sound level versus frequency;

providing second digital data representing an initial response curve of an initial hearing ability to be enhanced of sound level versus frequency;

comparing said first digital data to said second digital data and determining whether said initial response curve is within said tolerance range ; and

applying inherently said digital audio filters to digital signals representative of received sound to generate third digital data, converting said third digital data to an analog signal and providing said analog signal to a speaker in said hearing aid,

generating fourth digital data representative of an enhanced response curve of hearing ability of sound level versus frequency;

comparing said first digital data to said fourth digital data and determining whether said enhanced response curve is within said tolerance range; and

if said initial response curve is not within said tolerance range,

iteratively generating digital filter coefficients controlling center frequency, filter bandwidth and amplitude for a succession of additional digital audio filters to compensate said initial response curve,

automatically optimizing the center frequency, amplitude and filter bandwidth of said digital audio filters until said enhanced response curve is within said tolerance range or a predetermined limit on the number of digital audio filters has been reached, whichever occurs first (see US PAT. 6,721,428 claim 13 col. 14 lines 25-55).

The difference between the current claims and the patent is that the environments wherein the claimed digital filter in used the use of the claimed digital filter. The current application involve a hearing aid and the patent involves loudspeaker.

However, Both hearing aid and loudspeaker are involving an operation or turning of a speaker. This digital filter is meant to use with a speaker.

Although the environment (one is hearing aid, the other one is for loudspeaker) is different, but they both involve speakers. Therefore, using this digital filter in these two environments would have been obvious for one of ordinary skill in the art because even the environment changes, the operation of the digital filter remains substantially unchanged.

Consider claim 20 substantially all the claimed steps were claimed in the patent identified above, such as the steps of : " providing first digital data for N samples for a desired response curve of acceptable hearing ability of sound level versus frequency; providing second digital data representing N samples for an initial response curve of sound level versus frequency; and

generating total log-integral metric data according to the formula:

$$M = \sum_{i=1}^{N-1} \log_{10}\left(\frac{f_{i+1}}{f_i}\right)\left[\frac{\left|S(f_i)_{dB} - D(f_i)_{dB}\right| + \left|S(f_{i+1})_{dB} - D(f_{i+1})_{dB}\right|}{2}\right]$$

M is the total log-integral metric,

f is the frequency,

D is the first digital data,

S is the second digital data, and

N is the number of samples of first digital data and of second digital data (see US PAT. 6,721,428 claim15 and col.14 line 59-col.15 line15).

Claim Rejections - 35 USC § 103

4. The following is a quotation of 35 U.S.C. 103(a) which forms the basis for all obviousness rejections set forth in this Office action:

> (a) A patent may not be obtained though the invention is not identically disclosed or described as set forth in section 102 of this title, if the differences between the subject matter sought to be patented and the prior art are such that the subject matter as a whole would have been obvious at the time the invention was made to a person having ordinary skill in the art to which said subject matter pertains. Patentability shall not be negatived by the manner in which the invention was made.

5. Claims 1-9 and 14-19 are rejected under 35 U.S.C. 103(a) as being unpatentable over Gauthier (WO 90/09760) in view of Op de Beek (US PAT. 4,845,758).

Consider claim 1 Gauthier teaches a method for generating digital filter coefficients for corresponding digital filters for tuning a hearing aid employing digital audio processing to enhance hearing ability comprising:

providing first digital data (see fig.11 (target curve 118)) for a tolerance range for a target response curve representative of said enhanced hearing ability of sound level versus frequency;

providing second digital data (see fig.11 (loss curve 114)) representing an initial response curve of an initial hearing ability to be enhanced of sound level versus frequency;

comparing (see fig.10, (108)) said first digital data to said second digital data and determining whether said initial response curve is within said tolerance range (see page 34 line 15-page 35 line 5); but Gauthier does not clearly teach that if said initial response curve is not within said tolerance range, iteratively generating digital filter coefficients controlling center frequency, filter bandwidth and amplitude for a succession of additional digital audio filters, applying all currently generated digital audio filters to said second digital data to generate third digital data for a compensated response curve, and automatically optimizing the center frequency, amplitude and filter bandwidth of said currently generated digital audio filters until said compensated response curve is within said tolerance range or a predetermined limit on the number of digital audio filters has been reached, whichever occurs first.

However, Op de Beek teaches inherently that if said initial response curve is not within said tolerance range, iteratively generating digital filter coefficients controlling center frequency (see fig. 2a), filter bandwidth (the difference between the lowest and highest frequency and see fig.2a) and amplitude (gain and see fig.2b) for a succession of additional digital audio filters, applying all currently generated digital audio filters to

said second digital data to generate third digital data for a compensated response curve, and automatically optimizing the center frequency (see fig.2a), amplitude (gain and see fig.2b) and filter bandwidth (the difference between the lowest and highest frequency and see fig.2a) of said currently generated digital audio filters until said compensated response curve is within said tolerance range or a predetermined limit on the number of digital audio filters has been reached, whichever occurs first ((see figs. 5a-5d) and col.5 line 37-col.6 line 18).

Therefore, it would have obvious to one of ordinary skill in the art at the time the invention was made, to combine the teaching of Gauthier and Op De beek to provide the equalizer is characterized in that the central frequencies of at least those band filters whose bands are located in the low-frequency part of the frequency range are variable.

Consider claims 2-3, 5-6 Gauthier teaches a method of the step of iteratively generating digital audio filters is performed by iteratively generating second order filters (see fig.5 and page 15 line 10-15) and the initial response curve is an audiogram (see fig.11).

Consider claim 4 Gauthier teaches a method for generating a set of second order filter coefficients for corresponding digital filters to tune a hearing aid employing digital audio processing to enhance hearing ability comprising:

providing first digital data (see fig.11 (118 target curve)) for a tolerance range for a target response curve representative of said enhanced hearing ability of sound level versus frequency;

providing second digital data (see fig.11 (114 loss curve)) representative of an initial response curve of an initial hearing ability to be enhanced of sound level versus frequency;

comparing (see fig.10 (108)) said first digital data to said second digital data and determining whether said initial response curve is within said tolerance range (see page 34 line 15-page 35, line 5); but Gauthier does not clearly teach if said initial response curve is not within said tolerance range, generating a set of digital filter coefficients controlling center frequency, filter bandwidth and amplitude for a succession of digital audio filters to tune said hearing aid by performing the following optimizing steps iteratively,

digitally processing said second digital data to determine an n^{th} set of initial digital filter coefficients for an nth digital filter for an n^{th} peak in said actual initial curve where said initial response curve is not within said tolerance range, including a center frequency, and an amplitude and a bandwidth for said peak, where n^{th} is the number of an iteration of said optimizing steps,

digitally generating digital filter coefficients controlling center frequency, filter bandwidth and amplitude for a compensating n^{th} digital filter from said n^{th} set of initial parameters,

applying said n^{th} digital filter to said second digital data and modifying said n^{th} set of initial parameters to determine an n^{th} set of optimum digital filter coefficients for said compensating n^{th} digital filter, to generate third digital data for an n^{th} interim compensated response curve of sound level versus frequency,

processing said third digital data to determine whether said n^{th} interim compensated response curve is within said tolerance range,

if said n^{th} interim compensated response curve is not within said tolerance range, performing another iteration of said optimizing steps until said interim compensated response curve is within said tolerance range or a predetermined limit on the number of digital filters has been reached, whichever occurs first.

However, Op de Beek teaches if said initial response curve is not within said tolerance range, generating a set of digital filter coefficients controlling center frequency (see fig. 2a), filter bandwidth (the difference between the lowest and highest frequency and see fig.2a) and amplitude (gain and see fig.2b) for a succession of additional digital audio filters to tune said hearing aid by performing the following optimizing steps iteratively (see figs. 5a-5d) and col.5 line 37-col.6 line 18)),

digitally processing said second digital data to determine an n^{th} (see fig.1) set of initial digital filter coefficients for an nth digital filter (see fig.4) for an n^{th} peak in said actual initial curve where said initial response curve is not within said tolerance range, including center frequency (see fig. 2a), and amplitude (gain and see fig.2b) filter and bandwidth (the difference between the lowest and highest frequency and see fig.2a) for said peak, where n^{th} is the number of an iteration of said optimizing steps ((see figs. 5a-5d) and col.5 line 37-col.6 line 18),

digitally generating digital filter coefficients controlling center frequency (see fig. 2a), filter bandwidth (the difference between the lowest and highest frequency and see

fig.2a) and amplitude (gain and see fig.2b) for a compensating n^{th} digital filter from said n^{th} set of initial parameters (see figs. 5a-5b),

applying said n^{th} digital filter to said second digital data and modifying said n^{th} set of initial parameters to determine an n^{th} set of optimum digital filter coefficients for said compensating n^{th} digital filter, to generate third digital data for an n^{th} interim compensated response curve of sound level versus frequency ((see figs. 5a-5d) and figs. 7-8),

processing said third digital data to determine whether said n^{th} interim compensated response curve is within said tolerance range (see fig.2a),

if said n^{th} interim compensated response curve is not within said tolerance range, performing another iteration of said optimizing steps until said interim compensated response curve is within said tolerance range or a predetermined limit on the number of digital filters has been reached, whichever occurs first inherently (see figs. 5a-5d) and col.5 line 37-col.6 line 18)).

Therefore, it would have obvious to one of ordinary skill in the art at the time the invention was made, to combine the teaching of Gauthier and Op De beek to provide the equalizer is characterized in that the central frequencies of at least those band filters whose bands are located in the low-frequency part of the frequency range are variable.

Consider claim 7 Gauthier teaches a method for generating digital filter coefficients for corresponding digital filters for tuning a hearing aid employing digital audio processing to enhance hearing ability comprising:

providing first digital data (see fig.11 (118 target curve)) for a tolerance range for a target response curve representative of said enhanced hearing ability of sound level versus frequency;

providing second digital data (see fig.11 (114 loss curve)) for an initial response curve of said hearing ability to be enhanced of sound level versus frequency;

comparing (see fig.10 (108)) said first digital data to said second digital data and determining whether said initial response curve is within said tolerance range; and

applying said nth digital filter to said second digital data and modifying said n^{th} set of initial filter coefficients to determine an n^{th} set of optimum parameters for said n^{th} digital filter, to generate third digital data for an n^{th} interim compensated response curve of sound level versus frequency (see page 16 line 11-page 17 line 18);

if n > 1, performing the following joint filter optimizing steps iteratively and cyclically, generating fourth digital data for interim computed response curves in which for each joint filter optimizing iteration one of said n filters is absent, and then performing said single filter optimization steps utilizing said fourth digital data to generate fifth digital data for an updated interim response curve, digitally processing said fifth digital data to determine whether the most recent of said joint filter optimizing iterations has resulted in a change in said updated interim response curve greater than a predetermined amount of change, and if so continuing to perform said joint filter optimizing steps; processing said fifth digital data to determine whether said n^{th} interim compensated response curve is within said tolerance range, and if not (see page 35 line 20-page 36 line 24),

performing another iteration of the foregoing steps until said interim compensated response curve is within said tolerance range or a predetermined limit on the number of digital filters has been reached, whichever occurs first, but if so, ceasing performance of further iterations (see page 36 line 25-page 37 line 6),

but Gauthier does not clearly teach if said initial response curve is not within said tolerance range, generating a set digital filter coefficients controlling center frequency, filter bandwidth and amplitude for a succession of additional compensating digital audio filters by performing the following single filter optimizing steps iteratively, and digitally processing said second digital data to determine an n^{th} set of initial parameters for an n^{th} peak in said initial response curve where said initial response curve is not within said tolerance range, including a center frequency, an amplitude and a bandwidth for said peak, where n is the number of an iteration of said optimizing steps, digitally generating a compensating n^{th} digital filter from said n^{th} set of initial parameters.

However, Op de Beek teach if said initial response curve is not within said tolerance range, generating a set digital filter coefficients controlling center frequency (see fig. 2a), filter bandwidth (the difference between the lowest and highest frequency and see fig.2a) and amplitude (gain and see fig.2b) for a succession of additional compensating digital audio filters by performing the following single filter optimizing steps iteratively, and digitally processing said second digital data to determine an n^{th} set of initial parameters for an n^{th} peak in said initial response curve where said initial response curve is not within said tolerance range, including center frequency (see fig. 2a), filter

bandwidth (the difference between the lowest and highest frequency and see fig.2a) and amplitude (gain and see fig.2b) for said peak, where n is the number of an iteration of said optimizing steps, digitally generating a compensating n^{th} digital filter from said n^{th} set of initial parameters ((see figs. 5a-5d) and col.5 line 37-col.6 line 18).

Therefore, it would have obvious to one of ordinary skill in the art at the time the invention was made, to combine the teaching of Gauthier and Op De beek to provide the equalizer is characterized in that the central frequencies of at least those band filters whose bands are located in the low-frequency part of the frequency range are variable.

Consider claims 8-9 Gauthier teaches a method of the step of digitally generating a compensating n^{th} digital filter is performed by digitally generating a second-order filter (see fig.5 (516-528)); and the initial response curve is an audiogram (see fig.11).

Consider claims 12-13, Gauthier teaches the method of computer receives said first series of signals and indication signals generated by said device to generate said first digital date (see page 37 line 25-page 38 line 16); and first digital data is an audiogram (see fig.11 (118)).

Consider claim 14 Gauthier teaches an apparatus for generating digital filter coefficients for tuning a hearing aid digital audio processing for use by an individual, comprising:

a source of first audio digital data corresponding to analog audio signals having a selected frequency and multiple power levels; (see fig.1 (10));

Inherently, a digital to analog converter receiving (see fig.1,10 computer) said digital data from said digital audio processing unit and converting said processed digital

data into a corresponding analog audio signal; and a speaker receiving said analog audio signal from said digital to analog converter and producing corresponding sound to the individual (see page 6 lines 20-29);

a device for generating indication (16,20) signals indicative of said individual receiving said sound; and

a digital computer connected to receive said first audio digital data and said indication signals, said digital computer programmed to determine said digital filter coefficients for digital filters for tuning said hearing aid and provide said coefficients to said digital audio processing unit (see page 37 line 9-23).

But Gauther does not clearly teach a digital audio processing unit in said hearing aid for processing said first audio digital data according to at least one digital filter having digital filter coefficients controlling filter center frequency, amplitude and filter bandwidth and providing processed audio digital data, including applying digital audio filters for tuning said hearing aid characterized by coefficients in algorithms applied to said first audio digital data to effect said digital audio filters.

However, Op de Beek teaches a digital audio processing unit in said hearing aid for processing said first audio digital data according to at least one digital filter having digital filter coefficients controlling filter center frequency (see fig. 2a), filter bandwidth (the difference between the lowest and highest frequency and see fig.2a) and amplitude (gain and see fig.2b) and providing processed audio digital data, including applying digital audio filters for tuning said hearing aid characterized by coefficients in algorithms

applied to said first audio digital data to effect said digital audio filters ((see fig.5a-5d and col.5 line 37-col.6 line 18).

Therefore, it would have obvious to one of ordinary skill in the art at the time the invention was made, to combine the teaching of Gauthier and Op De beek to provide the equalizer is characterized in that the central frequencies of at least those band filters whose bands are located in the low-frequency part of the frequency range are variable.

Consider claim 15, Gauthier teaches that an apparatus of the digital computer is programmed to generate second digital data representative of said individual hearing ability when using said hearing aid without filters determined from said first audio digital data (see fig.5 (510) and page 16 line 11-page 17 line25) and said indication signals and to determine said coefficients by

providing third digital data (see fig.11, (118 target curve)) for a tolerance range for a target response curve of enhanced hearing of sound level versus frequency;

providing said second digital data (see fig.11 (114 loss curve)), wherein said second digital data represents an initial response curve of hearing ability of sound level versus frequency;

comparing (see fig.10 (108)) said third digital data to said second digital data and determining whether said initial response curve is within said tolerance range (see page 36 line 9-col. 37 line 6).

But Gauthier teaches if said initial response curve is not within said tolerance range,

iteratively generating digital filter coefficients controlling center frequency, filter bandwidth and amplitude for a succession of additional digital audio filters,

applying digital audio filters determined by said digital filter coefficients to said second digital data to generate fourth digital data for a compensated response curve, and automatically optimizing said digital filter coefficients by optimizing the center frequency, amplitude and filter bandwidth of said digital audio filters until said compensated response curve is within said tolerance range or a predetermined limit on the number of digital audio filters has been reached, whichever occurs first.

However Op de Beek teaches if said initial response curve is not within said tolerance range,

iteratively generating digital filter coefficients controlling center frequency (see fig. 2a), filter bandwidth (the difference between the lowest and highest frequency and see fig.2a) and amplitude (gain and see fig.2b) for a succession of additional digital audio filters,

applying digital audio filters determined by said digital filter coefficients to said second digital data to generate fourth digital data for a compensated response curve (see figs. 5a-5d and figs.7-8), and automatically optimizing said digital filter coefficients by optimizing the center frequency (see fig. 2a), filter bandwidth (the difference between the lowest and highest frequency and see fig.2a) and amplitude (gain and see fig.2b) of said digital audio filters until said compensated response curve is within said tolerance range or a predetermined limit on the number of digital audio filters has been reached, inherently whichever occurs first ((see figs. 5a-5d) and col.5 line 37-col.6 line 18) .

Therefore, it would have obvious to one of ordinary skill in the art at the time the invention was made, to combine the teaching of Gauthier and Op De beek to provide

the equalizer is characterized in that the central frequencies of at least those band filters whose bands are located in the low-frequency part of the frequency range are variable.

Consider claim 16 Gauthier teaches a method for generating digital filters for tuning a hearing aid to enhance hearing ability, comprising:

providing first digital data (see fig.11 (118 target curve)) for a tolerance range for a target response curve representative of said enhanced hearing ability of sound level versus frequency;

providing second digital data (see fig.11 (114 loss curve)) representing an initial response curve of an initial hearing ability to be enhanced of sound level versus frequency;

comparing (see fig.10 (108)) said first digital data to said second digital data and determining whether said initial response curve is within said tolerance range (see page 34 line 15-page 35 line 5); and

applying inherently said digital audio filters to digital signals representative of received sound to generate third digital data, converting said third digital data to an analog signal and providing said analog signal to a speaker in said hearing aid,

generating fourth digital data representative of an enhanced response curve of hearing ability of sound level versus frequency;

comparing said first digital data to said fourth digital data and determining whether said enhanced response curve is within said tolerance range (see page 34 line 15-page 35 line 5).

But Gauthier does not clearly teach if said initial response curve is not within said tolerance range,

iteratively generating digital filter coefficients controlling center frequency, filter bandwidth and amplitude for a succession of additional digital audio filters to compensate said initial response curve,

automatically optimizing the center frequency, amplitude and filter bandwidth of said digital audio filters until said enhanced response curve is within said tolerance range or a predetermined limit on the number of digital audio filters has been reached, whichever occurs first.

However, Op de Beek teaches if said initial response curve is not within said tolerance range,

iteratively generating digital filter coefficients controlling center frequency (see fig. 2a), filter bandwidth (the difference between the lowest and highest frequency and see fig.2a) and amplitude (gain and see fig.2b) for a succession of additional digital audio filters to compensate said initial response curve (see figs. 5a-5d and figs. 7-8),

automatically optimizing the center frequency (see fig. 2a), filter bandwidth (the difference between the lowest and highest frequency and see fig.2a) and amplitude (gain and see fig.2b) of said digital audio filters until said enhanced response curve is within said tolerance range or a predetermined limit on the number of digital audio filters has been reached, whichever inherently occurs first (see figs. 5a-5d) and col.5 line 37-col.6 line 18).

Therefore, it would have obvious to one of ordinary skill in the art at the time the invention was made, to combine the teaching of Gauthier and Op De beek to provide the equalizer is characterized in that the central frequencies of at least those band filters whose bands are located in the low-frequency part of the frequency range are variable.

Consider claims 17-19 Gauthier teaches a method of the step of iteratively generating digital audio filters is performed by iteratively generating second order filters (see page 35 line 20-page 36 line 24 (13 values is 13 digital filters); and the initial response curve is an audiogram (see fig.11 (loss curve)) and the enhanced response curve is an audiogram (see fig.11 (target curve)).

6. Claim 10 is rejected under 35 U.S.C. 103(a) as being unpatentable over Gauthier (WO 90/09760) in view of Sjursen (US PAT. 6,292,571).

Consider claim 10, Gauthier teaches a method for generating filters for tuning a hearing aid to enhance hearing ability of an individual comprising:

fitting said hearing aid to said individual (see abstract);

connecting (see fig.1) said hearing aid (26r,l) to a source of audio digital signals;

providing said individual with a device (16,20) to generate indication signals at will (see page 38 line 5-16);

receiving said indication (16,20) signal during said generation of a signal of a selected frequency indicative of said individual hearing said selected frequency (see page 38 line 18-26);

providing a digital computer (10) connected to receive said first series of audio digital signals (see fig.5 (510)) and said indication (16,20) signals to generate digital data representative of said individual's hearing ability using said hearing aid without filters determined from said first series of digital signals (see fig.5 (510) and see page 20 line 19-30)), said computer programmed to determine said digital filter coefficients for digital filters for tuning said hearing aid and providing said coefficients to said digital audio processing unit in said hearing aid (see page 37 line 7-23).

But Gauthier does not clearly teach generating and providing a first series of audio digital signals to said hearing aid, each digital signal in said first series corresponding to an analog audio signal having a selected frequency and multiple power levels;

at said hearing aid converting each of said series of digital signals into said corresponding analog audio signal;

providing a digital audio processing unit in said hearing aid for processing received audio digital signals corresponding to analog audio signals and providing processed audio digital data, including applying digital audio filters or tuning said hearing aid characterized by generating digital filter coefficients in algorithms applied to said received audio digital signals to effect said digital audio filters.

However, Sjursen teaches generating and providing a first series of audio digital signals to said hearing aid, each digital signal in said first series corresponding to an analog audio signal having a selected frequency and multiple power levels (see figs. 7-23);

at said hearing aid converting each of said series of digital signals into said corresponding analog audio signal (see col.4 lines 7-47);

providing a digital audio processing unit in said hearing aid for processing received audio digital signals corresponding to analog audio signals and providing processed audio digital data, including applying digital audio filters or tuning said hearing aid characterized by generating digital filter coefficients in algorithms applied to said received audio digital signals to effect said digital audio filters (see col.5 line 20-col.6 line 20).

Therefore, it would have obvious to one of ordinary skill in the art at the time the invention was made, to combine the teaching of Gauthier and Sjursen to provide A hearing aid using a digital filter where power consumption is minimized thus resulting in size advantage and cost economy.

7. Claims 11-13 are rejected under 35 U.S.C. 103(a) as being unpatentable over Gauthier (WO 90/09760) as modified by Sjursen (US PAT. 6,292,571) as applied to claim 10 above, and further in view of Op de Beek (US PAT. 4,845,758).

Consider claim 11 Gauthier teaches that a method of the digital computer is programmed to determine said digital filter coefficients by

providing second digital data (see fig.11 (118 target curve)) for a tolerance range for a target response curve ability of representative of said individual's enhanced hearing ability of sound level versus frequency;

providing first digital data (see fig.11 (114 loss curve)) representative of an initial response curve of said individual's hearing ability of sound level versus frequency;

comparing (see fig.10 (108)) said second digital data to said first digital data and determining whether said response curve is within said tolerance range (see page 34 line 15-page 35 line 50).

But Gauthier does not clearly teach if said response curve is not within said tolerance range, iteratively generating digital filter coefficients controlling center frequency, filter bandwidth and amplitude for a succession of additional digital audio filters,

applying digital audio filters determined by said digital filter coefficients to said first digital data to generate third digital data for a compensated response curve, and

automatically optimizing said digital filter coefficients by optimizing the center frequency, amplitude and filter bandwidth of said digital audio filters until said compensated response curve is within said tolerance range or a predetermined limit on the number of digital audio filters has been reached, whichever occurs firs.

However, Op de Beek teaches if said response curve is not within said tolerance range, iteratively generating digital filter coefficients controlling center frequency (see fig. 2a), filter bandwidth (the difference between the lowest and highest frequency and see fig.2a) and amplitude (gain and see fig.2b) for a succession of additional digital audio filters,

applying digital audio filters determined by said digital filter coefficients to said first digital data to generate third digital data for a compensated response curve(see figs5a-5d and figs 7-8), and

automatically optimizing said digital filter coefficients by optimizing the center frequency (see fig. 2a), amplitude (gain and see fig.2b) and filter bandwidth (the difference between the lowest and highest frequency and see fig.2a) of said digital audio filters until said compensated response curve is within said tolerance range or a predetermined limit on the number of digital audio filters has been reached, whichever inherently occurs first (see figs 5a-5d and col.5 line 37-col.6 line 18).

Therefore, it would have obvious to one of ordinary skill in the art at the time the invention was made, to combine the teaching of Gauthier and Sjiursen into Op De beek to provide the equalizer is characterized in that the central frequencies of at least those band filters whose bands are located in the low-frequency part of the frequency range are variable.

Consider claims 12-13, Gauthier teaches the method of computer receives said first series of signals and indication signals generated by said device to generate said first digital data (see page 37 line 25-page 38 line 16); and first digital data is an audiogram (see fig.11 (118)).

Response to Arguments

8. Applicant's arguments with respect to claims 1-20 have been considered but are moot in view of the new ground(s) of rejection.

Conclusion

9. Any response to this action should be mailed to:

Commissioner of Patents and Trademarks

Washington, D.C. 20231

or faxed to:(703) 872-9306

Hand-delivered responses should be brought to Crystal Park II, 2121 Crystal Drive, Arlington. VA., Sixth Floor (Receptionist).

Any inquiry conceming this communication or earlier communications from the examiner should be directed to Lao, Lun-See whose telephone number is (703) 305-2259 The examiner can normally be reached on Monday-Friday from 8:00 to 6:30. If attempts to reach the examiner by telephone are unsuccessful, the examiner's supervisor, Curtis Kuntz, can be reached on (703) 305-4708.

Any inquiry of a general nature or relating to the status of this application or proceeding should be directed to the Technology Center 2600 whose telephone number is (703) 306-0377.

Lao, Lun-See
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PRIMARY EXAMINER